Design 3D sound field with FIR Filters and IIR Filters

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1. INTRODUCTION

The dominant aim of this report is to describe how to simulate the three dimensional (3D) sound field using Finite Impulse Response (FIR) filters from the initial review 1 and Infinite Impulse Response (IIR) filters from Lab report 1 and 2.

1.1 Introductions and Problem Description

The physical effects of the diffraction of sound waves by the human torso, shoulders, head and pinna modify the spectrum of the sound that reaches the eardrums [1]. These chances can be simulated by Head Related Transfer Function (HRTF) and the same information can also be expressed in the time domain through the Head Related Impulse Response (HRIR). Using this impulse response can simulate 3D sound field by convolving a monaural signal for the two ears. And the result can be experienced by compensated headphones or through cross-talk-canceled stereo speakers. [2] While using HRIR is the most convincing way to simulate 3D sound field, it must be measured individually which is both time consuming and highly inconvenient. To solve this problem, several methods have proposed replacing measured HRIR to generate 3D sound field for each listeners in more simple and effective ways. For this purpose, Interaural Level Difference (ILD) and Interaural Time Differences (ITD), which caused by the presence of the human head introduced. They are the important cues to localize the sound source for humans. If we can rely on the assumption that the listener receives the sound material via a stereo headphone, we can reproduce most of the cues that are due to the filtering effect of the pinna-head-torso system, and inject the signal artificially affected by this filtering process directly to the ears. [3]

Interaural Time Differences (ITD) can be achieved by means of a first order allpass filter and also head shadowing effect can be attained by using low order IIR filters. These two cues, ITD and head shadowing effect, have been used in order to create the 3D sound field for listeners instead of using measured HRIR. This process has been proven through Lab report 2. However, since the pinna and shoulders provide multiple reflections that can be a strong cue to synthesize the sound field, adding the pinna reflections and shoulder echo using FIR filter models can create a stronger spatial sound field [4]. Combining an IIR head-shadowing model with a FIR pinna-echo model and a FIR shoulder-echo model is a simple and efficient way to replace measured HRIR in order to synthesize 3D sound field.

This report will be focused on how to implement FIR shoulder-echo model and FIR pinna-echo model in order to create modeled HRIR for 3D sound field. Also how to test the results in order to see how well these models represent HRIR compared to measured HRIR for 3D sound field.

2. SPECIFICATION

Since sounds can reach the pinna via two major paths such as diffraction around the head and reflection for the shoulders, combining an IIR head shadowing model with a FIR pinna-echo and a FIR shoulder-echo model is a simple and effective way to synthesize 3D sound field.

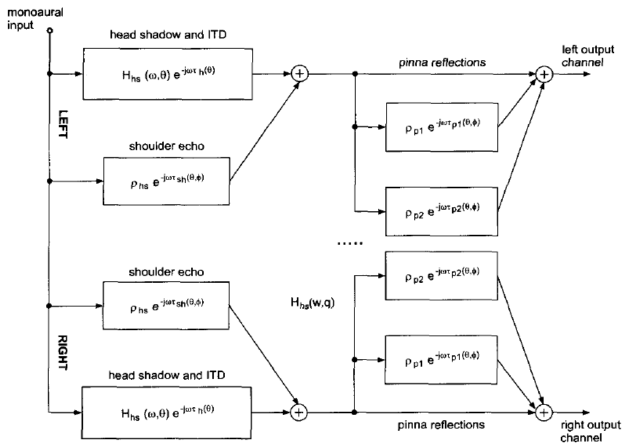

Figure 1. *A structural model of the head related transfer function. Separate modules account for head shadow, shoulder reflections, and pinna effects. Where $\rho's$ are reflection coefficients and the T's and t's are time delays.*

The human ears can discriminate interaural time difference that are as short as 10 μs therefore, it is important to control the time difference to simulate the 3D sound field [5]. Using simple IIR filters that provides the relative time delays can provide useful ITD cues. Interaural Time Difference (ITD) can be obtained by means of a first order allpass filter, which can be achieved by the equation below. [6]

$$ITD = \begin{cases} -\frac{a}{c}cos\theta & if\ 0 \leq |\theta| < \frac{\pi}{2} \\ \frac{a}{c}\left(|\theta| - \frac{\pi}{2}\right) & if\ \frac{\pi}{2} \leq |\theta|\pi \end{cases}$$

Equation 1. *The equation for calculating ITD. Where a=radius and c=speed of sound.*

The head shadowing effect can be explained by the loss of high frequencies and it occurs when the source is on the far side of the head. The effects can be simulated through the Matlab code called *headshadowingfilter* from the lab report 2. Single-zero head shadow filter can be calculated by the following equation.

$$H_{HS}(\omega,\theta) = \frac{1 + i\frac{\alpha\omega}{2\omega_0}}{1 + i\frac{\omega}{2\omega_0}}, 0 \leq \alpha(\theta)\ \leq 2$$

Equation 2. *Single-zero head shadow filter. α is the coefficient, which is a function of the angle of incidence θ, and controls the location of the zero.*

Where the frequency $\omega_0$is related to the radius of the sphere by $\omega_0 = \frac{c}{a}$, where c is speed of sound and a is the radius. If α=2, there is a 6dB boost at high frequencies, while if $\alpha < 1$, there is a cut [7].

As mentioned above, the time delay can be calculated by the equation 1. According to this equation, while the time delay is the smallest when the sound source locates near the right ear at 100°, the time delay is the largest when it locates left ear at -100°. Between 20° and 80°, there is a faint echo that arrives about 0.35ms after the main pulse. This time delay corresponds to a path length difference of 13cm, which is consistent with a shoulder reflection. The echo is most noticeable in the azimuth range due to the presence of the shoulders. Therefore, this fact can support that adding shoulder echo can generate better result. [8] And pinna reflections can be explained by the figure 2 below.

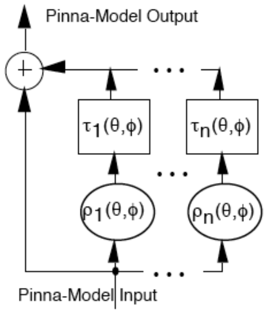

Figure 2. *Pinna model where ρ_1, ρ_2 ... are the amplitude T_1, T_1 .... T_n are the time delay.*

As shown above, the pinna reflections can be created by FIR filters with the simple adding and delaying process. When the sound source reaches human's ears, there are numerous numbers of reflections occur. Therefore, when the pinna-reflection model is implemented through FIR filters, the filters should be applied several times for the reflections.

3. IMPLEMENTATION

3.1 Interaural Time Difference (ITD) and Head Shadowing

In order to generate the head shadowing effect, using a first order allpass filter whose group delay is in seconds is the first step. The first order allpass filter and IIR filter can create ITD eventually.

As mentioned above, the head shadowing effect can be explained by the loss of high frequencies. This effect can cause the impulse response for the right ear to be large for azimuths near 90 degree and small for azimuths near -90 degree. To simulate the loss of high frequencies on the shadowed side, the impulse response will be filtered with the inverse of the head shadow filter given by equation 2. The overall magnitude and group delay responses of the block responsible for head shadowing and ITD. After obtained the filter and delay, produce the head-shadowing effect can be next step.

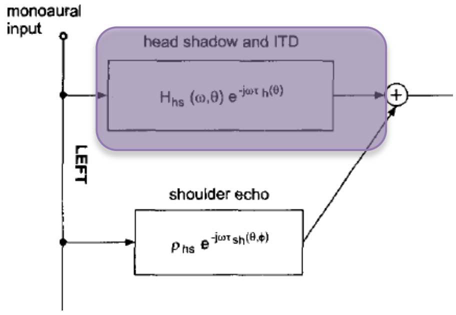

Figure 3. *Head shadowing and ITD can be obtained by using a function called headshadowingfilter from lab report 2 and this effect can be done by using an IIR filter.*

This effect can be done by using an IIR filter. All process can be done by using a function called *headshadowingfilter* from Lab report 2. The function has to be applied twice for left and right ears in order to place a sound at a certain position in virtual space. [9]

3.2 Shoulder echo and Pinna reflections

Basically, shoulder echo and pinna reflections can be implemented by using FIR filters. Since FIR filters are known as a non-recursive digital filter as they do not have the feedback loops. Therefore, FIR filter can be simply expressed with a simple delaying and adding process. Using the characteristic of FIR filter can implement shoulder echo and pinna reflections.

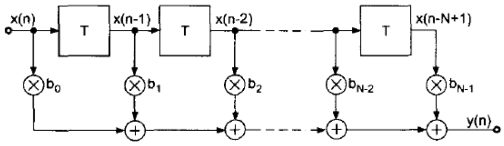

Figure 4. *Finite Impulse Response Filters. Where x (n-1), x (n-2)… are the time delay, b_0, b_1, b_2 …. are the gain differences.*

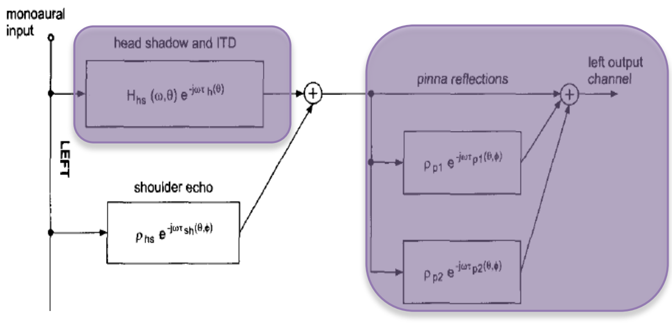

Figure 5. *Shoulder echo and Pinna reflections can be obtained by using FIR filters. Where ρ_{p1}, ρ_{p2} are the gain differences and $e^{-j\omega tp1\,(\theta,\emptyset)}, e^{-j\omega tp2\,(\theta,\emptyset)}$ are the time delay.*

The pinna provides multiples delays which can be obtained by means of a tapped delay line. Following formula provide a reasonable fit to the time delay for the *n*th pinna event. [10]

$$T_{pn}(\theta,\emptyset) = A_n \cos\left(\frac{\theta}{2}\right)\sin\left[D_n\left(90^\circ - \emptyset\right)\right] + B_n,$$
$$-90^\circ \leq \theta \leq 90^\circ, -90^\circ \leq \emptyset \leq 90^\circ$$

Equation 3. *An equation for the nth pinna event. Where A_n is an amplitude, B_n is an offset and $D_n$is a scaling factor. This equation is based on the measurement which were made at* $\theta = 0{,}15{,}30{,}45 \text{ and } 60^{\circ}$.

In order to implement FIR filters through the Matlab, *Filter* is an inbuilt function of Matlab can be used. To complete the Matlab function, filter coefficients are required. [11]

n	$\rho\, pn$	A_n[samples]	B_n[samples]	D_n
2	0.5	1	2	≅ 1
3	-1	5	4	≅ 0.5
4	0.5	5	7	≅ 0.5
5	-0.25	5	11	≅ 0.5
6	0.25	5	13	≅ 0.5

Table 1. *The filter coefficients for pinna reflections. Parameters for calculating amplitude and time delay of the reflections produced by the pinna model.*

When the sound source reaches human's ears, there are considerably large number of reflections will occur due to the presence the pinna. Therefore, a few number of FIR filters are required to simulate the pinna reflections.

The shoulder echo is synthesized in a single echo approximately. And the echo should also be attenuated as the source goes from frontal to lateral position. The time delay due to the presence of shoulders can be calculated by using the equation below. [12]

$$T_{sh} = 1.2\frac{180^{\circ} - \theta}{180^{\circ}}\left(1 - 0.00004\left((\emptyset - 80^{\circ})\frac{180^{\circ}}{180^{\circ} + \theta}\right)^2\right)$$

Equation 4. *The time delay due to the presence of shoulders in ms.*

In order to get FIR shoulder reflections model, Shoulder delay has to be calculated first which could be achieved with equation 4 in Matlab. A Matlab function called *headshadowingfilter* can be used to get the filter coefficients for shoulder reflections model.

a = (1 - shoulderdelay) / (1 + shoulderdelay); FIR filter coefficients can be calculated by using the Matlab code above. In order to implement FIR shoulder reflection filters through the Matlab, *Filter* is an inbuilt function of Matlab can be used. And those coefficients from the previous process will be used.

Through the Matlab, *Filter* function and filter coefficients from all the previous steps will be used. All these effects, head shadowing, shoulder echo and pinna reflections will be combined through the *Filter* function in Matlab. With this final process, the modeled HRIR can be achieved in order to simulate 3D sound field.

4. EVALUATION

To evaluate the models, all listening experiments will be performed in identical methods. All the experiments will be set up through Matlab. All the applicants will listen a white noise, which is filtered by either measured or modeled HRIR as a sound source for 3 seconds and played over the AKG-K77 stereo headphones. And there is a set of 8 reference sounds, which include the modeled HRIR sounds. First, the listener will listen the sound, which was filtered by his/her own measured HRIR. Then the listener will be asked to choose the sound which reference sound source is matched with what he/she is listening among the 8 reference sound sources. There are no limitations to the number of times the listeners can listen before making the final decision. This process will repeat with 10 different candidates. Second, the listener will listen the modeled HRIR sound source and be asked to choose the sound source among another set of 8 reference sound sources which are filtered by measured HRIR from different applicants and him/herself. The results from this experiment will indicate how well the modeled HRIR is corresponded with measured HRIR and will help to understand what to improve more.

There are several possibilities to allow the modeled HRIR can replace the measured HRIR, and the modeled HRIR will be adapted into several applications due to its simplicity and efficiency. This system allows the possibility of spatializing sound sources for binaural listening such as Erbe's Soundhack, award-winning software for the Macintosh, has a cookbook of HRTF coefficients and allow the musician to locate apparent sound sources in a 3D space. [13]

5. REFERENCE

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